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Adaptive end-to-end QoS for multimedia over heterogeneous wireless networks

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ABSTRACT

There has been a surge of interest for multimedia applications over wireless networks in recent years. A colossal number of ways have been proposed to decrease delay, delay jitter and loss in wireless networks and good user-perceived quality in video over internet. This paper studies the multimedia over heterogeneous wireless networks, requirements and problems, and proposes a new scheme to overcome the obstacles. The proposed scheme, takes into account the effects of Application-Level Wireless Multilevel ECN marking (AWMECN), thus helps us overcome the congestion/loss mistake problems. For handoff, handover and lossy link problems, it is considered that a freezing mechanism is in use in application layer and assumed that the upper layers can be aware of disconnection periods to make the rate adaptation decisions. Also a new scheme has been added to receiver to gracefully degrade the quality when no other action is available to combat the long delays without data which is caused by handoffs and wireless temporary disconnections. The transport layer mechanism is chosen to be UDP for avoiding TCP regarding problems. We believe that obtaining a good quality of video depends on good performance of all layers and tried to use the best mechanism in each layer.

1. Introduction

Internet is growing rapidly and its growth has been successful in the wired networks over the past years but the process for wireless and mobile networks is not as successful as for traditional networks. The story is going even worse when considering the heterogeneous wireless environment. Today's internet challenge is multimedia and QoS support, especially over wireless networks. A number of methods have been proposed to combat the problem over Internet's best-effort nature. New research trend is going to attack the problem in wireless and mobile networks.

There are three main impediments in QoS support, especially video streaming over the internet. First, the variable network performance due to load changes. More specifically available bandwidth is highly variable, which can be dealt with, to some extent, by using a buffer; but this option is limited by the buffer space available at the receiver. Second obstacle is the network congestion, which results in packet loss in network interior nodes and low arrival rate at the receiver. Congestion induced frame loss may severely affect video quality. Finally, the bandwidth availability in the network is highly unpredictable which makes quality adaptation difficult.

When one is to support QoS and more specifically video streaming over wireless networks, there is a need to go further to consider the problems of wireless and mobile networks as well. For an end-to-end QoS support, there always has been a choice of TCP or UDP. UDP is connection-less and does not maintain packet numbering and timing. It also does not provide any kind of information about the network state. There have been a number of ways proposing UDP in companion with RTP

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for time stamping and sequence number and using RTCP for obtaining network information to support QoS over wireless networks which is the preferred method for multimedia transmission over wireless networks [10]. The main advantage of this way is that UDP does not maintain packet loss and retransmission and it does not include any kind of congestion control mechanisms which take a lot of time, especially with loss/congestion mistake scenarios in wireless networks. But as the compression volume increase, the need to retain lost packets increase. On the other hand TCP works effectively over traditional networks but it does not work effectively for multimedia services as its performance degrades due to window size decrease in the response to packet loss resulting from the congestion.

TCP shows even worse performance over the wireless networks due to five main problems. First, limited bandwidth, which is a bottleneck to improve the TCP performance over wireless networks. Second, long round-trip times which cause the growth rate of the TCP congestion window to be very low resulting in low sending rates. Third, random losses due to low signal strength in an area or duration of noise; since TCP reacts to packet loss with halving its congestion window, bursts of errors may cause the TCP congestion window to reduce TCP sending rate dramatically. Fourth, User Mobility and handoffs; when mobile user transfers from a cell to another a handoff occurs due to user's need to change base station (BTS). Handoffs usually take about 300 ms and all packets get lost in this duration. Fifth, short flows services; this service is offered in a very short time interval. In the beginning of the connection TCP is still in slow-start and the data transmission may stop before the window size has enough time to increase sending rate.

There are number of ways to solve these problems including pure end-to-end protocols, link layer protocols, split connection protocols, soft-state transport-layer caching protocols, cross layer signaling protocols [8,10] but none of the proposed methods could solve all the TCP regarding problems in wireless networks.

In this paper we propose a way to overcome the video streaming problems in mobile wireless networks and provide an algorithm that shows better QoS and multimedia delivery over wireless networks. The proposed mechanism aims at adaptive end-to-end QoS support so it is classified in pure end-to-end protocols category.

The proposed mechanism tries to attack the QoS support problems with the main adaptation scheme used in application layer and a mixture of best services for each underlying layer. The main adaptation scheme which consists of rate and quality adaptation mechanisms and the graceful degrade mechanism are all done in application level. The Wireless MECN marking is used in network layer, but the decision about the MECN values is also made at application level. For the transport layer, it is assumed that the UDP is used and sequence numbering and time stamping is also done in application level header.

The rest of this paper is organized as follows: in Section 2 we study the problems encountered in QoS support over wireless networks and the ones we are going to solve in this paper. The third section describes in detail the proposed scheme. Simulation results are summarized in section four. Finally, section five includes conclusion remarks and a way to future work.

2. QoS support obstacles in wireless networks

We assumed the main problems of QoS support over wireless networks are random losses and handoffs as well as congestion. There are several proposed methods to overcome these problems [7,9]. The main problems and shortcomings of proposed features available till now are discussed here in detail. Then we study the multimedia characteristics and major obstacles in multimedia over heterogeneous wireless networks and show that how wireless disconnection durations affect the multimedia and its user-perceived quality over the mobile wireless networks.

2.1. Congestion

In the traditional TCP, packet loss is the main indicator of congestion events. In such networks, where the link error is very low and links are considered to be reliable, this mechanism would show acceptable results but when used in wireless networks, it may cause starting the congestion control mechanisms in the case of non-congestion related losses. In other words, random packet loss is one of the important obstacles in wireless networks that impose big bursts of packet loss in wireless networks. Therefore offering a good end-to-end QoS supporting scheme, requires considering an environment in which packets experience congestion related losses and also random losses due to wireless link. One of the problems to attack in this stage is to detect random losses from those originated from congestion events. One of the famous ways to solve this problem is explicit congestion notification. ECN [6,8,3] marking allows one to explicitly notify the receiver and therefore in ACK packets the sender, about the congestion and how to act accordingly. But for supporting QoS over the wireless networks it is not enough to have one bit indicator of congestion. A multi-level ECN mechanism as proposed in [2] best fits the requirements of multimedia applications that are the main group of applications needing QoS support. We used previously proposed method, Application Layer Wireless Multi-level ECN for obtaining a feedback from network backlog [3]. This multi-level ECN-like mechanism is done in the internal hops in the network and its worse value along the way is remained unchanged.

2.2. Handoff and temporary disconnections

The second problem in support of QoS in wireless networks is the handoff and temporary disconnection periods caused by user mobility. As mentioned earlier, in the mobile networks there is a disconnection gap while a mobile host moves from one